In the claims:

Amend claims 1-4, 17-19, 21, 25 and 28 as follows:

1.   (Currently Amended)   ~~An instantaneous encoding apparatus~~ Apparatus for analyzing an input sound signal, the apparatus comprising:

a unit signal generator for generating one or more unit signals, each of said unit signals having energy at a center frequency, and being represented by parameters including the center frequency and time variation rate of the center frequency;

an error calculator for calculating an error in an amplitude/phase space between the spectrum of said input sound signal and the spectrum of the sum of said one or more unit signals;

means for iteratively altering said parameters of the unit signals and for having said error calculator recalculate the error until the parameters of the unit signals that provide minimum error are determined; and

means for outputting as the encoded signals representing said input sound signal said one or more unit signals determined to provide the minimum error.

2.   (Currently Amended) The ~~instantaneous encoding~~ apparatus as claimed in claim 1, wherein said generator determines the number of unit signals

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to be generated responsive to the number of local peaks of power spectrum for said input signal.

3.   (Currently Amended) The ~~instantaneous encoding~~ apparatus as claimed in claim 1, wherein said center frequency corresponds to a local peak of the power spectrum for said input <u>sound</u> signal.

4.   (Currently Amended) The ~~instantaneous encoding~~ apparatus as claimed in claim 1, wherein said unit signal is represented by the equation:

$u(t)_i = a(t)_i \cos\left(2\pi \int f(t)_i dt\right)$ ($i = 1, 2, \ldots, k$) where $a(t)_i$ represents a time variation function for instantaneous amplitude and $f(t)_i$ a time variation function for instantaneous frequency.

5.   (Previously presented) A sound separation apparatus for separating a target signal from a mixed input signal, wherein the mixed input signal includes the target signal and one or more sound signals emitted from different sound sources, comprising:

a frequency analyzer for performing a frequency analysis on said mixed input signal to calculate a spectrum and for determining one or more frequency component candidate points at each time;

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feature extraction means for extracting feature parameters for said target signal, comprising:

a) a local layer for performing instantaneous encoding based on said spectrum to determine local feature parameters including frequencies, amplitudes and time variations thereof of the center frequencies of said frequency component candidate points;

b) a harmonic calculation layer for grouping the frequency component candidate points having a same harmonic structure that is determined by the local feature parameters including the frequency and its time variation rate of the frequency component candidate points, and then calculating a fundamental frequency of the harmonic structure, variations of the fundamental frequency, harmonics contained in the harmonic structure, and variations of the harmonics; and

c) a pitch continuity calculation layer for calculating a continuity of signal using the fundamental frequency and the variation of the fundamental frequency calculated by said harmonic calculation layer; and

a signal regenerator for regenerating a waveform of the target signal based on said feature parameters extracted by said feature extraction means.

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6. (Original) The sound separation apparatus as claimed in claim 5, wherein said local layer and global layers mutually supply the feature parameters analyzed in each layer to update the feature parameters in each layer based on said supplied feature parameters.

7. (Previously presented) The sound separation apparatus as claimed in claim 6, wherein said local layer is an instantaneous encoding layer for calculating frequencies, variations of said frequencies, amplitudes, and variations of said amplitudes at said frequency component candidate points.

8. (Cancelled)

9. (Original) The sound separation apparatus as claimed in claim 6, wherein said global layer further comprises a sound source direction prediction layer for predicting directions of sound sources for said mixed input signal.

10. (Original) The sound separation apparatus as claimed in claim 9, said global layer comprising:

a harmonic calculation layer for grouping frequency component candidate points having same harmonic structure based on said frequencies and the

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variations of frequency of said frequency component candidate points as well as the sound source directions predicted by the sound source direction prediction layer, and calculating a fundamental frequency of said harmonic structure, harmonics contained in said harmonic structure, and variation of the fundamental frequency and the harmonics; and

a pitch continuity calculation layer for calculating a continuity of signals using said fundamental frequency and said variation of the fundamental frequency at points of time.

11. (Previously presented) The sound separation apparatus as claimed in claim 5, wherein time variation rates are used as said variations.

12. (Original) The sound separation apparatus as claimed in claim 6, wherein each of said layers is logically composed of one or more computing elements, each computing elements being capable of calculating feature parameters, each computing elements mutually exchanging said calculated feature parameters with other elements included in upper and lower adjacent layers of one layer.

13. (Original) The sound separation apparatus as claimed in claim 12, said computing element executing steps comprising:

calculating a first consistency function indicating a degree of consistency between the feature parameters supplied from the computing element included in the upper adjacent layer and said calculated feature parameters,

calculating a second consistency function indicating a degree of consistency between the feature parameters supplied from the computing element included in the lower adjacent layer and said calculated feature parameters,

updating said feature parameters to maximize a validity indicator that is represented by a product of said first consistency function and said second consistency function.

14. (Original) The sound separation apparatus as claimed in claim 13, wherein said validity indicators are supplied to computing elements included in said lower adjacent layer.

15. (Original) The sound separation apparatus as claimed in claim 14, wherein a threshold value is calculated based on said supplied validity indicator and wherein said calculating element may be eliminated if the value of said validity indicator is less than said threshold value.

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16.   (Original) The sound separation apparatus as claimed in claim 14, wherein if the value of said validity indicator exceeds a given value, new computing elements are created in said lower layer.

17.   (Currently Amended)   An instantaneous encoding program recorded on a computer-readable medium and configured to execute the steps of:

performing a frequency analysis on an input signal to determine a spectrum;

generating one or more unit signals, each of said unit signals having energy at a center frequency, and being represented by parameters including the center frequency, time variation rate of the center frequency, amplitude of the center frequency and time variation rate of the amplitude;

calculating an error in amplitude/phase space between the spectrum of said input signal and the spectrum of the sum of said one or more unit signals;

iteratively altering said said parameters of the unit signals for iterative calculation of said error until the unit signals that provide minimum error are determined; and

outputting as the encoded signals representing said input signal said one or more unit signals determined to provide the minimum error.

18. (Original) The instantaneous encoding program as claimed in claim 17, wherein said generating step includes determining the number of unit signals to be generated responsive to the number of local peaks of power spectrum for said input signal.

19. (Previously presented) The instantaneous encoding program as claimed in claim 17, wherein a frequency is selected from local peaks of power spectrum for said input signal.

20. (Previously presented) The instantaneous encoding program as claimed in claim 17, wherein said parameters are modeled by a function.

21. (Currently Amended) A sound separation program recorded on a computer-readable medium and configured for separating a target sound signal from a mixed input sound signal, wherein the mixed input signal includes the target sound signal and one or more sound signals emitted from different sound sources, the program being configured to execute the steps of:

performing a frequency analysis on said mixed input sound signal to calculate a spectrum and determine one or more frequency component candidate points at each time;

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extracting feature parameters <u>related to</u> said target <u>sound</u> signal including:

a) determining at a local layer local feature parameters based on said spectrum at said frequency component candidate points, said local feature parameters including one or more center frequencies, time variation rate of the center frequencies to analyze local feature parameters;

b) a harmonic calculation layer for grouping the unit signals having a same harmonic structure that is determined by the local feature parameters including the time variation rate of the center frequencies, and then calculating a fundamental frequency of the harmonic structure, variations of the fundamental frequency, harmonics contained in the harmonic structure, and variations of the harmonics; and

c) calculating at a pitch continuity calculation layer a continuity of signal based on said fundamental frequency and said variation of the fundamental frequency at each point in time; and

regenerating a waveform of the target <u>sound</u> signal based on said feature parameters extracted by the extracting step.

22. (Original) The sound separation program as claimed in claim 21, wherein said local layer and global layers mutually supply the feature parameters

analyzed in each layer to update the feature parameters in each layer based on said supplied feature parameters.

23. (Original) The sound separation program as claimed in claim 22, wherein said local layer is an instantaneous encoding layer for calculating frequencies, variations of said frequencies, amplitudes, and variations of said amplitudes for said frequency component candidate points.

24. (Cancelled)

25. (Currently amended) The sound separation program as claimed in claim 22, wherein said global layer further comprises a sound source direction prediction layer for predicting directions of sound sources for said mixed input sound signal.

26. (Original) The sound separation program as claimed in claim 25, said global layer comprising:

a harmonic calculation layer for grouping frequency component candidate points having same harmonic structure based on said frequencies and the variations of frequency of said frequency component candidate points as well as

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the sound source directions predicted by the sound source direction prediction

layer, and calculating a fundamental frequency of said harmonic structure,

harmonics contained in said harmonic structure, and variation of the fundamental

frequency and the harmonics; and

a pitch continuity calculation layer for calculating a continuity of signals

using said fundamental frequency and said variation of the fundamental frequency

at points of time.

27.   (Previously presented)   The sound separation program as claimed in

claim 21, wherein time variation rates are used as said variations.

28.   (Currently amended) A sound separation program as claimed in

claim 22, wherein each of said layers are logically composed of one or more

computing elements, each computing ~~elements~~ <u>element</u> being capable of

calculating feature parameters, each computing ~~elements~~ <u>element</u> mutually

exchanging said calculated feature parameters with other <u>computing</u> elements

included in upper and lower adjacent layers of one layer.

29.   (Original)   The sound separation program as claimed in claim 28,

said computing element executing steps comprising:

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calculating a first consistency function indicating a degree of consistency between the feature parameters supplied from the computing element included in the upper adjacent layer and said calculated feature parameters,

calculating a second consistency function indicating a degree of consistency between the feature parameters supplied from the computing element included in the lower adjacent layer and said calculated feature parameters,

updating said feature parameters to maximize a validity indicator that is represented by a product of said first consistency function and said second consistency function.

30.   (Original)   The sound separation program as claimed in claim 29, wherein said validity indicators are supplied to computing elements included in said lower adjacent layer.

31.   (Original)   The sound separation program as claimed in claim 30, wherein a threshold value is calculated based on said supplied validity indicator and wherein said calculating element may be eliminated if the value of said validity indicator is less than said threshold value.

32. (Original) The sound separation program as claimed in claim 30, wherein if the value of said validity indicator exceeds a given value, new computing elements are created in said lower layer.